IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re the Application of:

Nils Peter Nordqvist, et al.

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For: HEARING PROSTHESIS WITH AUTOMATIC CLASSIFICATION OF THE LISTENING ENVIRONMENT

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Examiner: Ensey, Brian

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P.O. Box 1450
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CORRESPONDENCE

Dear Sir:

This correspondence is in response to the Office Communication mailed on 11/21/07. According to the Office Communication, claim 67 is to be amended so that it depends from claim "61" (instead of claim 73). However, Applicant respectfully notes that claim 67 recites "the plurality of discrete Hidden Markov Models," which is supported by claim 73. Thus, Applicant respectfully requests that the patent be issued in accordance with the original claim 67 (which the Office Communication indicates is renumbered as claim 39). A listing of the allowed claims is attached for the Examiner's convenience.

If the Examiner has any questions or comments regarding this correspondence, the Examiner is respectfully requested to contact the undersigned at (650) 849-4960.

Respectfully submitted,

Dated: 1/8/08

By: [signature]
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LISTING OF ALLOWED CLAIMS

28. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment;

an output transducer for converting a processed output signal into an electrical or an acoustic output signal;

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal;

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm;

the processing means being further adapted to:

segment the input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames,

compare each of the feature vectors, $O(t)$, with a feature vector set to determine, for substantially each feature vector, an associated symbol value so as to generate an observation sequence of symbol values associated with the consecutive signal frames, wherein the feature vector set has been determined in an off-line training procedure which utilized real-life sound source recordings made through an input signal path of a target hearing prosthesis or by performing a substantially similar signal processing of an input signal to simulate characteristics of the input signal path, and stored in non-volatile memory locations of the hearing prosthesis,

process the observation sequence of symbol values with at least one discrete Hidden Markov Model, $\lambda^{source} = \{A^{source}, B^{source}, \alpha_0^{source}\}$, associated with a predetermined sound source to determine element value(s) of a classification vector indicating a probability of the predetermined sound source being active in the listening environment,

control one or several values of the related algorithm parameters in dependence of the element value(s) of the classification vector,

thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment, wherein

A^{source} = A state transition probability matrix,

B^{source} = An observation symbol probability distribution matrix for an input observation $O(t)$ for each state of the at least one Hidden Markov Model, and

α_0^{source} = An initial state probability distribution vector.

29. A hearing prosthesis according to claim 28, wherein the processing means are adapted to process the observation sequence of symbol values with a plurality of discrete Hidden Markov Models associated with respective predetermined sound sources to determine the element values of the classification vector indicating a probability of each predetermined sound source.

30. A hearing prosthesis according to claim 28, wherein the feature vectors are associated with respective integer symbol values during a vector quantization process.

31. A hearing prosthesis according to claim 28, wherein the feature vector set comprises between 8 and 256 discrete symbols.

32. A hearing prosthesis according to claim 29, wherein the processing means further comprises a decision controller adapted to smooth inherent time scales of the plurality of discrete

Hidden Markov Models by monitoring element values of the classification vector and control the one or several values of the related algorithm parameters.

33. A hearing prosthesis according to claim 32, wherein the decision controller comprises a Hidden Markov Model operating on a substantially longer time scale of the input signal than the inherent time scales of the plurality of discrete Hidden Markov Models.

34. A hearing prosthesis according to claim 32, wherein the inherent time scales of the plurality of discrete Hidden Markov Models are selected within a range of 10-100 milliseconds and the substantially longer time scale of the Hidden Markov Model is selected within a range of 1-60 seconds.

35. A hearing prosthesis according to claim 28, wherein the predetermined sound source is constituted by a mixture of speech and/or traffic noise and/or babble noise mixed together in a predetermined proportion.

36. A hearing prosthesis according to claim 28, wherein the predetermined sound source is a mixture of speech and babble noise with a particular target signal to noise ration.

37. A hearing prosthesis according to claim 28, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

38. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment;

an output transducer for converting a processed output signal into an electrical or an acoustic output signal;

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal;

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm;

the processing means being further adapted to:

segment the input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames,

compare each of the feature vectors, $O(t)$, with a feature vector set to determine, for substantially each feature vector, an associated symbol value so as to generate an observation sequence of symbol values associated with the consecutive signal frames,

process the observation sequence of symbol values with a plurality of discrete Hidden Markov Models, $\lambda^{source} = \{A^{source}, B^{source}, \alpha_0^{source}\}$, associated with respective predetermined sound sources to determine element values of a classification vector indicating a probability of each predetermined sound source being active in the listening environment, wherein the processing means further comprises a decision controller adapted to smooth inherent time scales of the plurality of discrete Hidden Markov Models by monitoring element values of the classification vector and control the one or several values of the related algorithm parameters,

control one or several values of the related algorithm parameters in dependence of the element value(s) of the classification vector,

thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment, wherein

A^{source} = A state transition probability matrix,

B^{source} = An observation symbol probability distribution matrix for an input observation $O(t)$ for each state of the at least one Hidden Markov Model, and

α_0^{source} = An initial state probability distribution vector.

39. A hearing prosthesis according to claim 38, wherein the feature vectors are associated with respective integer symbol values during a vector quantization process.

41. A hearing prosthesis according to claim 38, wherein the feature vector set comprises between 8 and 256 discrete symbols.

42. A hearing prosthesis according to claim 38, wherein the feature vector set has been determined in an off-line training procedure which utilized real-life sound source recordings and stored in non-volatile memory locations of the hearing instrument.

43. A hearing prosthesis according to claim 38, wherein the decision controller comprises a Hidden Markov Model operating on a substantially longer time scale of the input signal than the inherent time scales of the plurality of discrete Hidden Markov Models.

44. A hearing prosthesis according to claim 38, wherein the inherent time scales of the plurality of discrete Hidden Markov Models are selected within a range of 10-100 milliseconds and the substantially longer time scale of the Hidden Markov Model is selected within a range of 1-60 seconds.

45. A hearing prosthesis according to claim 38, wherein the predetermined sound source is constituted by a mixture of speech and/or traffic noise and/or babble noise mixed together in a predetermined proportion.

46. A hearing prosthesis according to claim 38, wherein the predetermined sound source is a mixture of speech and babble noise with a particular target signal to noise ration.

47. A hearing prosthesis according to claim 38, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

48. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment;

an output transducer for converting a processed output signal into an electrical or an acoustic output signal;

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal;

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm;

the processing means being further adapted to:

segment the input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames, wherein the value of T_{frame} lies between 1 and 100 milliseconds,

compare each of the feature vectors, $O(t)$, with a feature vector set to determine, for substantially each feature vector, an associated symbol value so as to generate an observation sequence of symbol values associated with the consecutive signal frames,

process the observation sequence of symbol values with at least one discrete Hidden Markov Model, $\lambda^{source} = \{A^{source}, B^{source}, \alpha_0^{source}\}$, associated with a predetermined sound source to determine element value(s) of a classification vector indicating a probability of the predetermined sound source being active in the listening environment,

control one or several values of the related algorithm parameters in dependence of the element value(s) of the classification vector,

thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment, wherein

A^{source} = A state transition probability matrix,

B^{source} = An observation symbol probability distribution matrix for an input observation $O(t)$ for each state of the at least one Hidden Markov Model, and

α_0^{source} = An initial state probability distribution vector.

49. A hearing prosthesis according to claim 48, wherein the value of T_{frame} lies between 5 and 10 milliseconds.

50. A hearing prosthesis according to claim 48, wherein the processing means are adapted to process the observation sequence of symbol values with a plurality of discrete Hidden Markov Models associated with respective predetermined sound sources to determine the element values of the classification vector indicating a probability of each predetermined sound source.

51. A hearing prosthesis according to claim 48, wherein the feature vectors are associated with respective integer symbol values during a vector quantization process.

52. A hearing prosthesis according to claim 48, wherein the feature vector set comprises between 8 and 256 discrete symbols.

53. A hearing prosthesis according to claim 48, wherein the feature vector set has been determined in an off-line training procedure which utilized real-life sound source recordings and stored in non-volatile memory locations of the hearing instrument

54. A hearing prosthesis according to claim 53, wherein the real-life sound recordings have been made through an input signal path of a target hearing prosthesis or by performing a substantially similar signal processing of an input signal to simulate characteristics of the input signal path.

55. A hearing prosthesis according to claim 50, wherein the processing means further comprises a decision controller adapted to smooth inherent time scales of the plurality of discrete Hidden Markov Models by monitoring element values of the classification vector and control the one or several values of the related algorithm parameters.

56. A hearing prosthesis according to claim 55, wherein the decision controller comprises a Hidden Markov Model operating on a substantially longer time scale of the input signal than the inherent time scales of the plurality of discrete Hidden Markov Models.

57. A hearing prosthesis according to claim 55, wherein the inherent time scales of the plurality of discrete Hidden Markov Models are selected within a range of 10-100 milliseconds and the substantially longer time scale of the Hidden Markov Model is selected within a range of 1-60 seconds.

58. A hearing prosthesis according to claim 48, wherein the predetermined sound source is constituted by a mixture of speech and/or traffic noise and/or babble noise mixed together in a predetermined proportion.

59. A hearing prosthesis according to claim 48, wherein the predetermined sound source is a mixture of speech and babble noise with a particular target signal to noise ration.

60. A hearing prosthesis according to claim 48, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

61. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment;

an output transducer for converting a processed output signal into an electrical or an acoustic output signal;

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal;

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm;

the processing means being further adapted to:

segment the input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames,

compare each of the feature vectors, $O(t)$, with a feature vector set to determine, for substantially each feature vector, an associated symbol value so as to generate an observation sequence of symbol values associated with the consecutive signal frames,

process the observation sequence of symbol values with at least one ergodic Hidden Markov Model, $\lambda^{source} = \{A^{source}, B^{source}, \alpha_0^{source}\}$, associated with a predetermined sound source to determine element value(s) of a classification vector indicating a probability of the predetermined sound source being active in the listening environment,

control one or several values of the related algorithm parameters in dependence of the element value(s) of the classification vector,

thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment, wherein

A^{source} = A state transition probability matrix,

B^{source} = An observation symbol probability distribution matrix for an input observation, and $O(t)$ for each state of the at least one Hidden Markov Model

α_0^{source} = An initial state probability distribution vector.

62. A hearing prosthesis according to claim 61, wherein the processing means are adapted to process the observation sequence of symbol values with a plurality of discrete Hidden Markov Models associated with respective predetermined sound sources to determine the element values of the classification vector indicating a probability of each predetermined sound source.

63. A hearing prosthesis according to claim 61, wherein the feature vectors are associated with respective integer symbol values during a vector quantization process.

64. A hearing prosthesis according to claim 61, wherein the feature vector set comprises between 8 and 256 discrete symbols.

65. A hearing prosthesis according to claim 61, wherein the feature vector set has been determined in an off-line training procedure which utilized real-life sound source recordings and stored in non-volatile memory locations of the hearing instrument

66. A hearing prosthesis according to claim 65, wherein the real-life sound recordings have been made through an input signal path of a target hearing prosthesis or by performing a substantially similar signal processing of an input signal to simulate characteristics of the input signal path.

67. A hearing prosthesis according to claim 61, wherein the processing means further comprises a decision controller adapted to smooth inherent time scales of the plurality of discrete Hidden Markov Models by monitoring element values of the classification vector and control the one or several values of the related algorithm parameters.

68. A hearing prosthesis according to claim 67, wherein the decision controller comprises a Hidden Markov Model operating on a substantially longer time scale of the input signal than the inherent time scales of the plurality of discrete Hidden Markov Models.

69. A hearing prosthesis according to claim 67, wherein the inherent time scales of the plurality of discrete Hidden Markov Models are selected within a range of 10-100 milliseconds and the substantially longer time scale of the Hidden Markov Model is selected within a range of 1-60 seconds.

70. A hearing prosthesis according to claim 61, wherein the predetermined sound source is constituted by a mixture of speech and/or traffic noise and/or babble noise mixed together in a predetermined proportion.

71. A hearing prosthesis according to claim 61, wherein the predetermined sound source is a mixture of speech and babble noise with a particular target signal to noise ration.

72. A hearing prosthesis according to claim 61, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

73. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment;

an output transducer for converting a processed output signal into an electrical or an acoustic output signal;

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal;

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm;

the processing means being further adapted to:

segment the input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames,

compare each of the feature vectors, $O(t)$, with a feature vector set to determine, for substantially each feature vector, an associated symbol value so as to generate an observation sequence of symbol values associated with the consecutive signal frames,

process the observation sequence of symbol values with at least one discrete Hidden Markov Model, $\lambda^{source} = \{A^{source}, B^{source}, \alpha_0^{source}\}$, associated with a predetermined sound source to determine element value(s) of a classification vector indicating a probability of the predetermined sound source being active in the listening environment, wherein the predetermined sound source is constituted by a mixture of speech and/or traffic noise and/or babble noise mixed together in a predetermined proportion,

control one or several values of the related algorithm parameters in dependence of the element value(s) of the classification vector,

thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment, wherein

A^{source} = A state transition probability matrix,

B^{source} = An observation symbol probability distribution matrix for an input observation $O(t)$ for each state of the at least one Hidden Markov Model, and

α_0^{source} = An initial state probability distribution vector.

74. A hearing prosthesis according to claim 73, wherein the processing means are adapted to process the observation sequence of symbol values with a plurality of discrete Hidden Markov Models associated with respective predetermined sound sources to determine the element values of the classification vector indicating a probability of each predetermined sound source.

75. A hearing prosthesis according to claim 73, wherein the feature vectors are associated with respective integer symbol values during a vector quantization process.

76. A hearing prosthesis according to claim 73, wherein the feature vector set comprises between 8 and 256 discrete symbols.

77. A hearing prosthesis according to claim 73, wherein the feature vector set has been determined in an off-line training procedure which utilized real-life sound source recordings and stored in non-volatile memory locations of the hearing instrument

78. A hearing prosthesis according to claim 77, wherein the real-life sound recordings have been made through an input signal path of a target hearing prosthesis or by performing a substantially similar signal processing of an input signal to simulate characteristics of the input signal path.

79. A hearing prosthesis according to claim 74, wherein the processing means further comprises a decision controller adapted to smooth inherent time scales of the plurality of discrete Hidden Markov Models by monitoring element values of the classification vector and control the one or several values of the related algorithm parameters.

80. A hearing prosthesis according to claim 79, wherein the decision controller comprises a Hidden Markov Model operating on a substantially longer time scale of the input signal than the inherent time scales of the plurality of discrete Hidden Markov Models.

81. A hearing prosthesis according to claim 79, wherein the inherent time scales of the plurality of discrete Hidden Markov Models are selected within a range of 10-100 milliseconds and the substantially longer time scale of the Hidden Markov Model is selected within a range of 1-60 seconds.

82. A hearing prosthesis according to claim 73, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

83. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment;

an output transducer for converting a processed output signal into an electrical or an acoustic output signal;

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal;

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm;

the processing means being further adapted to:

segment the input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames,

compare each of the feature vectors, $O(t)$, with a feature vector set to determine, for substantially each feature vector, an associated symbol value so as to generate an observation sequence of symbol values associated with the consecutive signal frames,

process the observation sequence of symbol values with at least one discrete Hidden Markov Model, $\lambda^{source} = \{A^{source}, B^{source}, \alpha_0^{source}\}$, associated with a predetermined sound source to determine element value(s) of a classification vector indicating a probability of the predetermined

sound source being active in the listening environment, wherein the predetermined sound source is a mixture of speech and babble noise with a particular target signal to noise ration,

control one or several values of the related algorithm parameters in dependence of the element value(s) of the classification vector,

thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment, wherein

A^{source} = A state transition probability matrix,

B^{source} = An observation symbol probability distribution matrix for an input observation $O(t)$ for each state of the at least one Hidden Markov Model, and

α_0^{source} = An initial state probability distribution vector.

84. A hearing prosthesis according to claim 83, wherein the processing means are adapted to process the observation sequence of symbol values with a plurality of discrete Hidden Markov Models associated with respective predetermined sound sources to determine the element values of the classification vector indicating a probability of each predetermined sound source.

85. A hearing prosthesis according to claim 83, wherein the feature vectors are associated with respective integer symbol values during a vector quantization process.

86. A hearing prosthesis according to claim 83, wherein the feature vector set comprises between 8 and 256 discrete symbols.

87. A hearing prosthesis according to claim 83, wherein the feature vector set has been determined in an off-line training procedure which utilized real-life sound source recordings and stored in non-volatile memory locations of the hearing instrument

88. A hearing prosthesis according to claim 87, wherein the real-life sound recordings have been made through an input signal path of a target hearing prosthesis or by performing a substantially similar signal processing of an input signal to simulate characteristics of the input signal path.

89. A hearing prosthesis according to claim 84, wherein the processing means further comprises a decision controller adapted to smooth inherent time scales of the plurality of discrete Hidden Markov Models by monitoring element values of the classification vector and control the one or several values of the related algorithm parameters.

90. A hearing prosthesis according to claim 89, wherein the decision controller comprises a Hidden Markov Model operating on a substantially longer time scale of the input signal than the inherent time scales of the plurality of discrete Hidden Markov Models.

91. A hearing prosthesis according to claim 89, wherein the inherent time scales of the plurality of discrete Hidden Markov Models are selected within a range of 10-100 milliseconds and the substantially longer time scale of the Hidden Markov Model is selected within a range of 1-60 seconds.

92. A hearing prosthesis according to claim 83, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

93. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment;

an output transducer for converting a processed output signal into an electrical or an acoustic output signal;

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal;

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm;

the processing means being further adapted to:

segment the input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames,

compare each of the feature vectors, $O(t)$, with a feature vector set to determine, for substantially each feature vector, an associated symbol value so as to generate an observation sequence of symbol values associated with the consecutive signal frames,

process the observation sequence of symbol values with at least one discrete Hidden Markov Model, $\lambda^{source} = \{A^{source}, B^{source}, \alpha_0^{source}\}$, associated with a predetermined sound source to determine element value(s) of a classification vector indicating a probability of the predetermined sound source being active in the listening environment,

control one or several values of the related algorithm parameters in dependence of the element value(s) of the classification vector,

thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment, wherein

A^{source} = A state transition probability matrix,

B^{source} = An observation symbol probability distribution matrix for an input observation $O(t)$ for each state of the at least one Hidden Markov Model, and

α_0^{source} = An initial state probability distribution vector.

94. A hearing prosthesis according to claim 93, wherein the processing means are adapted to process the observation sequence of symbol values with a plurality of discrete Hidden Markov Models associated with respective predetermined sound sources to determine the element values of the classification vector indicating a probability of each predetermined sound source.

95. A hearing prosthesis according to claim 93, wherein the feature vectors are associated with respective integer symbol values during a vector quantization process.

96. A hearing prosthesis according to claim 93, wherein the feature vector set comprises between 8 and 256 discrete symbols.

97. A hearing prosthesis according to claim 93, wherein the feature vector set has been determined in an off-line training procedure which utilized real-life sound source recordings and stored in non-volatile memory locations of the hearing instrument

98. A hearing prosthesis according to claim 97, wherein the real-life sound recordings have been made through an input signal path of a target hearing prosthesis or by performing a substantially similar signal processing of an input signal to simulate characteristics of the input signal path.

99. A hearing prosthesis according to claim 94, wherein the processing means further comprises a decision controller adapted to smooth inherent time scales of the plurality of discrete Hidden Markov Models by monitoring element values of the classification vector and control the one or several values of the related algorithm parameters.

100. A hearing prosthesis according to claim 99, wherein the decision controller comprises a Hidden Markov Model operating on a substantially longer time scale of the input signal than the inherent time scales of the plurality of discrete Hidden Markov Models.

101. A hearing prosthesis according to claim 99, wherein the inherent time scales of the plurality of discrete Hidden Markov Models are selected within a range of 10-100 milliseconds and the substantially longer time scale of the Hidden Markov Model is selected within a range of 1-60 seconds.

102. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer for converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal,

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm,

the processing means being further adapted to:

segment the input signal into consecutive signal frames of time duration T^{frame}, and generate respective feature vectors, *O(t)*, representing predetermined signal features of the consecutive signal frames, wherein the value of T_{frame} lies between 1 and 100 milliseconds,

process the feature vectors with one or more Hidden Markov Models operating on a first time scale and associated with respective predetermined sound sources to determine element values of a first classification vector indicating a probability of the predetermined sound sources being active in the listening environment,

process the first classification vector with a Hidden Markov Model operating at a second time scale and associated with one or more predetermined sound sources to determine element values of the classification vector,

control one or several values of the related algorithm parameters in dependence of element values of the classification vector,

thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment.

103. A hearing prosthesis according to claim 102, wherein the value of T_{frame} lies between 5 and 10 milliseconds.

104. A hearing prosthesis according to claim 102, wherein the first time scale is selected within the range 10-100 milliseconds, and the second time scale is selected within the range 1-60 seconds.

105. A hearing prosthesis according to claim 102, wherein the one or more Hidden Markov Models comprises between 2 and 10 states.

106. A hearing prosthesis according to claim 102, wherein the predetermined sound sources are constituted by a mixture of speech and/or traffic noise and/or babble noise mixed together in a predetermined proportion.

107. A hearing prosthesis according to claim 102, wherein the predetermined sound sources are mixtures of speech and babble noise with a particular target signal to noise ration.

108. A hearing prosthesis according to claim 102, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

109. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer for converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal,

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm,

the processing means being further adapted to:

segment the input signal into consecutive signal frames of time duration T^{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames,

process the feature vectors with the one or more Hidden Markov Models operating on a first time scale and associated with respective predetermined sound sources to determine element values of a first classification vector indicating a probability of the predetermined sound sources being active in the listening environment,

process the first classification vector with a Hidden Markov Model operating at a second time scale and associated with one or more predetermined sound sources to determine element values of the classification vector, wherein the first time scale is selected within the range 10-100 ms and the second time scale is selected within the range 1-60 seconds,

control one or several values of the related algorithm parameters in dependence of element values of the classification vector,

thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment.

110. A hearing prosthesis according to claim 109, wherein the one or more Hidden Markov Models comprises between 2 and 10 states.

111. A hearing prosthesis according to claim 109, wherein the predetermined sound sources are constituted by a mixture of speech and/or traffic noise and/or babble noise mixed together in a predetermined proportion.

112. A hearing prosthesis according to claim 109, wherein the predetermined sound sources are mixtures of speech and babble noise with a particular target signal to noise ration.

113. A hearing prosthesis according to claim 109, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

114. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer for converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal,

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm,

the processing means being further adapted to:

segment the input signal into consecutive signal frames of time duration T^{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames,

process the feature vectors with the one or more ergodic Hidden Markov Models operating on a first time scale and associated with respective predetermined sound sources to determine element values of a first classification vector indicating a probability of the predetermined sound sources being active in the listening environment,

process the first classification vector with a Hidden Markov Model operating at a second time scale and associated with one or more predetermined sound sources to determine element values of the classification vector,

control one or several values of the related algorithm parameters in dependence of element values of the classification vector,

thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment.

115. A hearing prosthesis according to claim 114, wherein the first time scale is selected within the range 10-100 milliseconds, and the second time scale is selected within the range 1-60 seconds.

116. A hearing prosthesis according to claim 114, wherein the predetermined sound sources are constituted by a mixture of speech and/or traffic noise and/or babble noise mixed together in a predetermined proportion.

117. A hearing prosthesis according to claim 114, wherein the predetermined sound sources are mixtures of speech and babble noise with a particular target signal to noise ration.

118. A hearing prosthesis according to claim 114, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

119. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer for converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal,

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm,

the processing means being further adapted to:

segment the input signal into consecutive signal frames of time duration T^{frame}, and generate respective feature vectors, *O(t)*, representing predetermined signal features of the consecutive signal frames,

process the feature vectors with the one or more Hidden Markov Models operating on a first time scale and associated with respective predetermined sound sources to determine element values of a first classification vector indicating a probability of the predetermined sound sources being active in the listening environment,

process the first classification vector with a Hidden Markov Model operating at a second time scale and associated with one or more predetermined sound sources to determine element values of the classification vector, wherein the predetermined sound sources are constituted by a mixture of speech and/or traffic noise and/or babble noise mixed together in a predetermined proportion,

control one or several values of the related algorithm parameters in dependence of element values of the classification vector,

thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment.

120. A hearing prosthesis according to claim 119, wherein the first time scale is selected within the range 10-100 milliseconds, and the second time scale is selected within the range 1-60 seconds.

121. A hearing prosthesis according to claim 119, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

122. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer for converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal,

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm,

the processing means being further adapted to:

segment the input signal into consecutive signal frames of time duration T^{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames,

process the feature vectors with the one or more Hidden Markov Models operating on a first time scale and associated with respective predetermined sound sources to determine element values of a first classification vector indicating a probability of the predetermined sound sources being active in the listening environment,

process the first classification vector with a Hidden Markov Model operating at a second time scale and associated with one or more predetermined sound sources to determine element

values of the classification vector, wherein the predetermined sound sources are mixtures of speech and babble noise with a particular target signal to noise ration,

control one or several values of the related algorithm parameters in dependence of element values of the classification vector,

thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment.

123. A hearing prosthesis according to claim 122, wherein the first time scale is selected within the range 10-100 milliseconds, and the second time scale is selected within the range 1-60 seconds.

124. A hearing prosthesis according to claim 122, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

125. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer for converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal,

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm,

the processing means being further adapted to:

segment the input signal into consecutive signal frames of time duration T^{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames,

process the feature vectors with the one or more Hidden Markov Models operating on a first time scale and associated with respective predetermined sound sources to determine element values of a first classification vector indicating a probability of the predetermined sound sources being active in the listening environment,

process the first classification vector with a Hidden Markov Model operating at a second time scale and associated with one or more predetermined sound sources to determine element values of the classification vector, wherein the predetermined sound sources are mixtures of speech and babble noise with a particular target signal to noise ration,

control one or several values of the related algorithm parameters in dependence of element values of the classification vector,

thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment.

126. A hearing prosthesis according to claim 125, wherein the first time scale is selected within the range 10-100 milliseconds, and the second time scale is selected within the range 1-60 seconds.

127. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer for converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with at least two predetermined signal processing algorithms and respective sets of algorithm parameters to generate the processed output signal,

a memory area storing values of the respective algorithm parameters for the at least two predetermined signal processing algorithms,

the processing means being further adapted to:

segment an input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames, wherein the value of T_{frame} lies between 1 and 100 milliseconds,

process the feature vectors with at least one Hidden Markov Model $\lambda^{source} = \{A^{source}, b(O(t)), \alpha_0^{source}\}$, associated with a predetermined sound source to determine element values of a classification vector indicating a probability of the predetermined sound source being active in the listening environment,

control a transition between the at least two predetermined signal processing algorithms in dependence of element values of the classification vector, wherein:

A^{source} = A state probability matrix,

$b(O(t))$ = Probability function for an input observation O(t) for each state of the at least one Hidden Markov Model, and

α_0^{source} = An initial state probability distribution vector.

128. A hearing prosthesis according to claim 127, wherein the value of T_{frame} lies between 5 and 10 milliseconds.

129. A hearing prosthesis according to claim 127, comprising a pair of omni-directional microphones generating a pair of input signals to provide the hearing prosthesis with a directional signal mode and a non-directional signal mode and wherein the processing means control the transition between the directional and non-directional signal mode.

130. A hearing prosthesis according to claim 127, wherein the predetermined sound source is constituted by a mixture of speech and/or traffic noise and/or babble noise mixed together in a predetermined proportion.

131. A hearing prosthesis according to claim 127, wherein the predetermined sound source is a mixture of speech and babble noise with a particular target signal to noise ration.

132. A hearing prosthesis according to claim 127, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

133. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer for converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with at least two predetermined signal processing algorithms and respective sets of algorithm parameters to generate the processed output signal,

a memory area storing values of the respective algorithm parameters for the at least two predetermined signal processing algorithms,

the processing means being further adapted to:

segment an input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames,

process the feature vectors with at least one ergodic Hidden Markov Model $\lambda^{source} = \{A^{source}, b(O(t)), \alpha_0^{source}\}$, associated with a predetermined sound source to determine element values of a classification vector indicating a probability of the predetermined sound source being active in the listening environment,

control a transition between the at least two predetermined signal processing algorithms in dependence of element values of the classification vector, wherein:

A^{source} = A state probability matrix,

$b(O(t))$ = Probability function for an input observation O(t) for each state of the at least one Hidden Markov Model, and

α_0^{source} = An initial state probability distribution vector.

134. A hearing prosthesis according to claim 133, comprising a pair of omni-directional microphones generating a pair of input signals to provide the hearing prosthesis with a directional

signal mode and a non-directional signal mode and wherein the processing means control the transition between the directional and non-directional signal mode.

135. A hearing prosthesis according to claim 133, wherein the predetermined sound source is constituted by a mixture of speech and/or traffic noise and/or babble noise mixed together in a predetermined proportion.

136. A hearing prosthesis according to claim 133, wherein the predetermined sound source is a mixture of speech and babble noise with a particular target signal to noise ration.

137. A hearing prosthesis according to claim 133, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

138. A hearing prosthesis comprising:

a pair of omni-directional microphones adapted to generate a pair of input signals in response to receiving an acoustic signal from a listening environment to provide the hearing prosthesis with a directional signal mode and a non-directional signal mode,

an output transducer for converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the pair of input signals in accordance with a respective pair of predetermined signal processing algorithms and respective sets of algorithm parameters to generate the processed output signal,

a memory area storing values of the respective algorithm parameters for the at least two predetermined signal processing algorithms,

the processing means being further adapted to:

segment an input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames,

process the feature vectors with at least one ergodic Hidden Markov Model $\lambda^{source} = \{A^{source}, b(O(t)), \alpha_0^{source}\}$, associated with a predetermined sound source to determine element values of a classification vector indicating a probability of the predetermined sound source being active in the listening environment,

control a transition between the at least two predetermined signal processing algorithms in dependence of element values of the classification vector, wherein:

A^{source} = A state probability matrix,

$b(O(t))$ = Probability function for an input observation O(t) for each state of the at least one Hidden Markov Model, and

α_0^{source} = An initial state probability distribution vector.

139. A hearing prosthesis according to claim 138, wherein the predetermined sound source is constituted by a mixture of speech and/or traffic noise and/or babble noise mixed together in a predetermined proportion.

140. A hearing prosthesis according to claim 138, wherein the predetermined sound source is a mixture of speech and babble noise with a particular target signal to noise ration.

141. A hearing prosthesis according to claim 138, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

142. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer for converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with at least two predetermined signal processing algorithms and respective sets of algorithm parameters to generate the processed output signal,

a memory area storing values of the respective algorithm parameters for the at least two predetermined signal processing algorithms,

the processing means being further adapted to:

segment an input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames,

process the feature vectors with at least one Hidden Markov Model $\lambda^{source} = \{A^{source}, b(O(t)), \alpha_0^{source}\}$, associated with a predetermined sound source to determine element values of a classification vector indicating a probability of the predetermined sound source being active in the listening environment, wherein the predetermined sound source is constituted by a mixture of speech and/or traffic noise and/or babble noise mixed together in a predetermined proportion,

control a transition between the at least two predetermined signal processing algorithms in dependence of element values of the classification vector, wherein:

A^{source} = A state probability matrix,

$b(O(t))$ = Probability function for an input observation O(t) for each state of the at least one Hidden Markov Model, and

α_0^{source} = An initial state probability distribution vector.

143. A hearing prosthesis according to claim 142, comprising a pair of omni-directional microphones generating a pair of input signals to provide the hearing prosthesis with a directional signal mode and a non-directional signal mode and wherein the processing means control the transition between the directional and non-directional signal mode.

144. A hearing prosthesis according to claim 142, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

145. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer for converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with at least two predetermined signal processing algorithms and respective sets of algorithm parameters to generate the processed output signal,

a memory area storing values of the respective algorithm parameters for the at least two predetermined signal processing algorithms,

the processing means being further adapted to:

segment an input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames,

process the feature vectors with at least one Hidden Markov Model $\lambda^{source} = \{A^{source}, b(O(t)), \alpha_0^{source}\}$, associated with a predetermined sound source to determine element values of a classification vector indicating a probability of the predetermined sound source being active in the listening environment, wherein the predetermined sound source is a mixture of speech and babble noise with a particular target signal to noise ration,

control a transition between the at least two predetermined signal processing algorithms in dependence of element values of the classification vector, wherein:

A^{source} = A state probability matrix,

$b(O(t))$ = Probability function for an input observation O(t) for each state of the at least one Hidden Markov Model, and

α_0^{source} = An initial state probability distribution vector.

146. A hearing prosthesis according to claim 145, comprising a pair of omni-directional microphones generating a pair of input signals to provide the hearing prosthesis with a directional signal mode and a non-directional signal mode and wherein the processing means control the transition between the directional and non-directional signal mode.

147. A hearing prosthesis according to claim 145, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames.

148. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer for converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with at least two predetermined signal processing algorithms and respective sets of algorithm parameters to generate the processed output signal,

a memory area storing values of the respective algorithm parameters for the at least two predetermined signal processing algorithms,

the processing means being further adapted to:

segment an input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames,

process the feature vectors with at least one Hidden Markov Model $\lambda^{source} = \{A^{source}, b(O(t)), \alpha_0^{source}\}$, associated with a predetermined sound source to determine element values of a classification vector indicating a probability of the predetermined sound source being active in the listening environment,

control a transition between the at least two predetermined signal processing algorithms in dependence of element values of the classification vector, wherein:

A^{source} = A state probability matrix,

$b(O(t))$ = Probability function for an input observation O(t) for each state of the at least one Hidden Markov Model, and

α_0^{source} = An initial state probability distribution vector.

149. A hearing prosthesis according to claim 148, comprising a pair of omni-directional microphones generating a pair of input signals to provide the hearing prosthesis with a directional signal mode and a non-directional signal mode and wherein the processing means control the transition between the directional and non-directional signal mode.

150. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer fro converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal,

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm,

the processing means being further adapted to:

segment an input signal into consecutive signal frames of time duration, T_{frame} and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames, wherein the value of T_{frame} lies between 1 and 100 milliseconds,

process the feature vectors with a set of Hidden Markov Models modeling respective isolated words or commands to determine element values of a classification vector indicating a probability of an isolated word or command being spoken,

thereby making the hearing prosthesis capable of recognizing a corresponding set of isolated words or commands.

151. A hearing prosthesis according to claim 150, wherein the value of T_{frame} lies between 5 and 10 milliseconds.

152. A hearing prosthesis according to claim 150, wherein the processing means is adapted to recognize voice commands from the user to control one or several functions of the hearing prosthesis.

153. A hearing prosthesis according to claim 150, wherein the set of Hidden Markov Models utilizes left-right Hidden Markov Models.

154. A hearing prosthesis according to claim 150, wherein a training of the set of Hidden Markov Models has been performed on words or commands spoken by the user during a fitting session.

155. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer fro converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal,

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm,

the processing means being further adapted to:

segment an input signal into consecutive signal frames of time duration, T_{frame} and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames,

process the feature vectors with a set of ergodic Hidden Markov Models modeling respective isolated words or commands to determine element values of a classification vector indicating a probability of an isolated word or command being spoken,

thereby making the hearing prosthesis capable of recognizing a corresponding set of isolated words or commands.

156. A hearing prosthesis according to claim 155, wherein the processing means is adapted to recognize voice commands from the user to control one or several functions of the hearing prosthesis.

157. A hearing prosthesis according to claim 155, wherein the set of Hidden Markov Models utilizes left-right Hidden Markov Models.

158. A hearing prosthesis according to claim 155, wherein a training of the set of Hidden Markov Models has been performed on words or commands spoken by the user during a fitting session.

159. A hearing prosthesis comprising:

a microphone adapted to generate an input signal in response to receiving an acoustic signal from a listening environment,

an output transducer fro converting a processed output signal into an electrical or an acoustic output signal,

processing means adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal,

a memory area storing values of the related algorithm parameters for the predetermined signal processing algorithm,

the processing means being further adapted to:

segment an input signal into consecutive signal frames of time duration, T_{frame} and generate respective feature vectors, $O(t)$, representing predetermined signal features of the consecutive signal frames, wherein each of the feature vectors comprises a plurality of cepstrum parameters or differential cepstrum parameters representing the predetermined signal features of the consecutive signal frames,

process the feature vectors with a set of Hidden Markov Models modeling respective isolated words or commands to determine element values of a classification vector indicating a probability of an isolated word or command being spoken,

thereby making the hearing prosthesis capable of recognizing a corresponding set of isolated words or commands.

160. A hearing prosthesis according to claim 159, wherein the processing means is adapted to recognize voice commands from the user to control one or several functions of the hearing prosthesis.

161. A hearing prosthesis according to claim 159, wherein the set of Hidden Markov Models utilizes left-right Hidden Markov Models.

162. A hearing prosthesis according to claim 159, wherein a training of the set of Hidden Markov Models has been performed on words or commands spoken by the user during a fitting session.